AMENDMENTS TO THE CLAIMS

This listing of claims will replace all prior versions and listings of claims in the application:

LISTING OF CLAIMS:

1. (canceled).

2. (currently amended): ~~The method according to claim 1,~~ A method of improving sound playback of digitized speech signals transmitted to a telecommunications terminal at the beginning of a telephone call set up over a communications network where the signals are transmitted in the form of packets in the event said call is set up from a sending telecommunications terminal comprising voice activity detection means for transmitting only those digitized sound signal packets that contain speech signals, said digitized sound signal packets that contain speech signals are taken from a set of sound signal packets that are available for being transmitted after the sound has been digitized and encoded in the sending terminal, the method comprising transmitting sound signal packets from the digitizing and encoding means without taking account of the presence or absence of speech signals in the processed sound signals during an initial stage of call optimization,

wherein the initial telephone call optimization stage during which digitized sound signal packets are transmitted from a sending terminal without taking account of the presence or absence of speech signals in the processed signals is of a duration that is selected in such a manner as to enable a receiving terminal to receive a sufficient number of digitized sound signal packets relating to the call to enable the size of ~~the~~ a receive buffer for digitized sound signal

packets to be determined on the basis of a statistical evaluation of the delays observed on the

received packets.

3. (currently amended): Telecommunications hardware which is connected to a network

enabling packets to be exchanged and which is designed to communicate over the network with a

compatible terminal by means of digitized sound signal packets including digitized speech

signals produced in the context of a VOIP type telephone call that is set up over the network

under IP protocol or an equivalent protocol, the hardware comprising:

a programmed control unit enabling a number of digitized sound signal packets to be

transmitted when a telephone call is set up and during an initial optimization ~~stage, said number~~

~~being sufficient to enable~~ stage having a duration that enables a receiver terminal to receive a

sufficient number of the digitized sound signal packets relating to the call to determine the size

of a receive buffer for the received digitized sound signal packets by statistically evaluating the

delays observed on the received packets; and

voice activity determining means enabling digitized sound signals to be transmitted only

if said digitized sound signals contain speech signals, said voice activity determining means

being prevented from acting until the initial optimization stage has terminated.

4. (previously presented): Telecommunications hardware according to claim 3, wherein

said programmed control unit comprises a timing means acting on the voice activity determining

means so that said voice activity determining means act only after the end of an initial

optimization stage of determined duration.

5. (currently amended): Telecommunications hardware according to claim 4, wherein the timing means ~~act~~ acts to temporarily inhibit the action of the voice activity determining means until the end of the initial stage of call optimization.

6. (currently amended): The method according to claim ~~1,~~ 2, wherein the telephone call set up is a VOIP call set up under Internet protocol.

7. (currently amended): Telecommunication hardware according to claim 3, wherein ~~particular~~ the telecommunication hardware is a subscriber terminal or a common terminal.